ALTEC LANSING ENGINEERING NOTES

TECHNICAL LETTER NO. 232A

Equalization Techniques and Practices

by Ted Uzzle

Introduction

Acoustic equalization, as we are accustomed to do it with real time analysers and contiguous-band filter sets, has two main goals: increasing the gain of a sound system, and improving the naturalness of its perceived sound quality. While the principles of equalization are widely used in audio, these two applications account for most of its use, and require careful understanding.

Equalization and System Gain

There are two electro-acoustic limitations on sound system gain: the destruction of the transducers (in the case of all sound systems), and feedback (when there is regeneration present in the system).

In multi-way systems each group of similar loudspeakers has an L_{max} above which it will destroy itself; that is to say, the woofers will blow up at a certain level, the tweeters will blow up at a certain level, and so on. In well-designed systems all these maximum levels will be within a few decibels of each other, but this is not always possible, particularly in long-throw outdoor applications where a thirty-foot stack of woofers might be required to achieve this uniformity. Equalization can adjust these relative levels so that maximum output is available when needed.

Within an individual transducer there are several destruction modes, the two most common being thermal failure and mechanical failure. These will usually be characteristic of overcurrents in the voice coils in different frequency bands (although this is not a simple relationship). The very experienced loudspeaker user can adjust the spectrum of each transducer to approximate the L_{max} possible in each equalization band.

A far more common limitation on system gain is feedback in sound systems where the microphone and the loudspeaker are in the same room. A sound system will feed back at the first frequency to rise to the feedback ceiling. If that first frequency is prominent in the feedforward frequency response, then by equalizing the feedforward frequency response to reduce the prominent frequency, all other frequencies will be able to operate at a higher level before the system does feed back. This additional increment in frequency bands less prone to feedback represents additional W_L, or loudspeaker acoustic power output, available and usable from the system.

Natural Sound Quality

Few topics suffer the effects of more preposterous claims and delusions than the use of equalization to achieve naturalness. Let's try to dredge up some common sense from the morass.

Naturalness in sound reproduction is a complex combination of accuracy, familiarity, comfort, illusion, and other factors. There is no cookbook for achieving it, and may never be.

Do not make one criterion paramount and satisfy it at the expense of all others. Attempting a frequency response ruler—flat from barometric pressure to gamma rays may seem intellectually satisfying, but it sounds awful.

Remember that different loudspeakers sound most natural with different frequency responses. This is an especially important consideration when an attempt is made to extend the bandwidth of the loudspeakers, with equalization. Remember also that microphones and real time analysers display a frequency response integrated over time, and this is not how the ears perceive sound.

Do not try to adjust for hearing losses. In the nineteenth century it was fashionable for art historians to explain the sinuous, elongated figures in El Greco's visionary paintings optically: the poor man suffered from astigmatism! How different his work would have been had he been able to buy spectacles! Of course, he used the same eyes to look at his canvasses and to look at his models; in the same way, those with imperfect hearing use the same ears to listen to real speech and music, as reproduced sounds.

Be certain you are attempting to equalize only the sounds originating in the loudspeakers, and not ambient noise or rattles in the room excited by the loudspeakers. Remember also that vibrations can be transmitted up the microphone stand and their harmonics created in the floor or the stand can appear in the analyser display.

The final test for naturalness is an experienced listener who is familiar with the sound equipment in many similar installations.

Spectrum Shaping

Many years of experience with sound system equalization show that a high-frequency rolloff should be applied. Figure 1 shows a recommended acoustical frequency response curve for speech reinforcement systems. As with all the curves shown here, it should be tailored to the individual installation, because of unique factors involving air absorption at high frequencies, variation in directivity index of the loudspeaker system and the test microphone over frequency, the room's reverberant characteristics, and other factors.

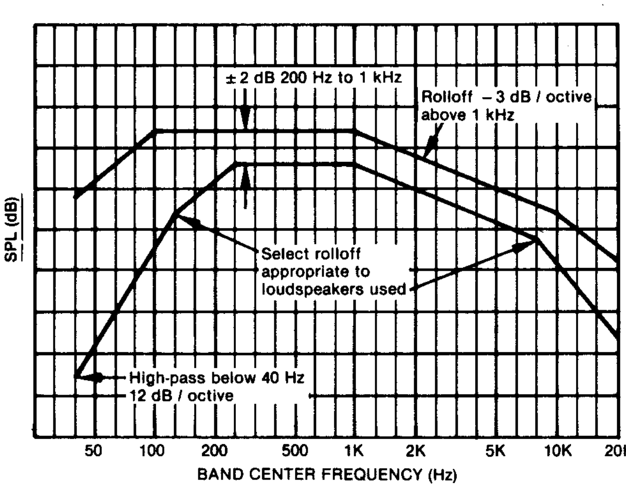

Figure 1. Recommended response curve for speech reinforcement systems.

Figure 2 shows the acoustic response characteristic now embodied in an international standard for motion picture sound playback in the theatre.

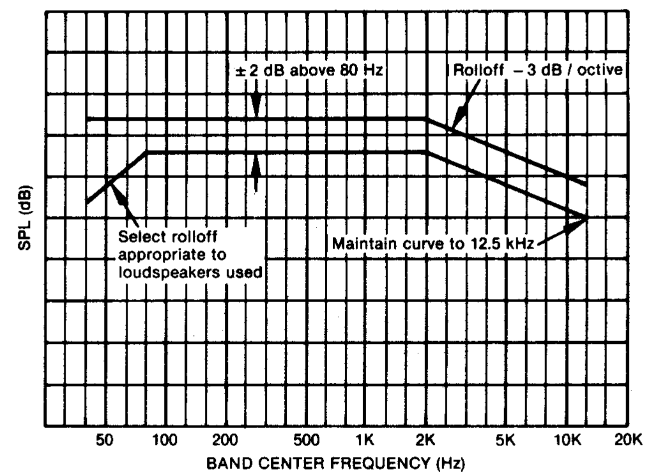

Figure 2. International standard response curve for cinema playback systems.

Figure 3 shows the acoustic response of monitoring and control rooms in recording studios. The frequency extremes are adjusted for the particular loudspeakers used and the preference of the recording engineers.

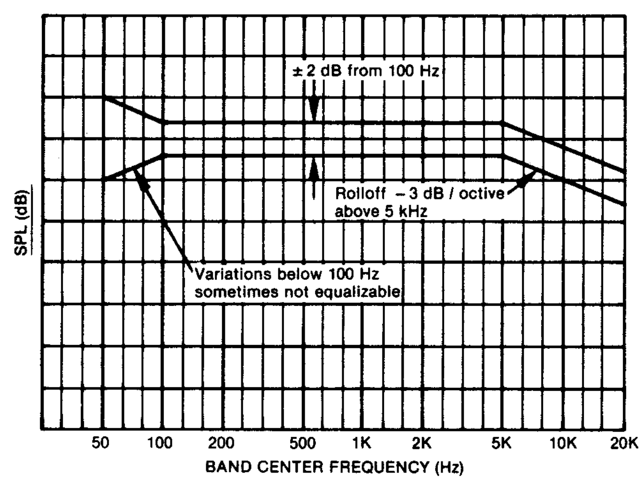

Figure 3. Recommended response curve for studio control room monitoring systems.

Figure 4 shows a typical high-level rock music reinforcement system.

All these contours are typical of modern practice, but must be considered as suggestions only, to be modified after careful listening and experienced judgment.

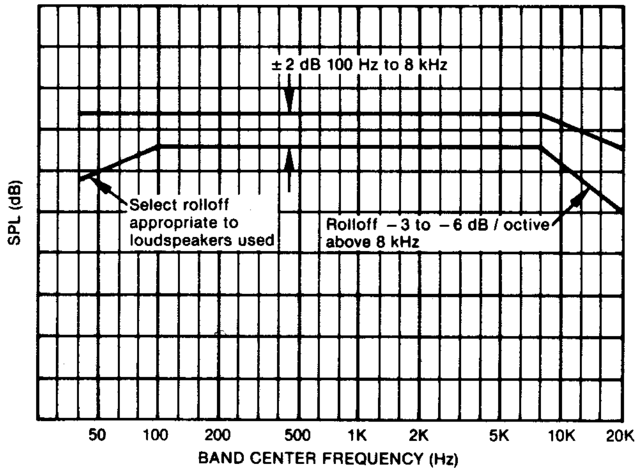

Figure 4. Recommended response curve for high-level rock music reinforcement systems.

Equalization Technique

There are as many different ways to achieve good equalizations as there are sound engineers achieving them. These suggestions are made as a brief checklist which may propose useful individual techniques to those performing equalizations.

Before attempting to equalize, talk through a speech reinforcement system or play music or film through a playback system, with an exeprienced listener present. Play a slow sine-wave sweep. If the speech and music sound distorted, or if the sine-wave sweep exposes rattles, frequencies with a harsh, guttural quality, or sharp, sudden changes in sound quality, stop and correct these problems before attempting the equalization. They can be made worse by equalization, and they can invalidate the apparent response seen on the screen of the real time analyser.

Before attempting to equalize, walk through the room and watch the response in the 2 kHz—4 kHz band. This level search may suggest correction of the coverage; whether or not this correction can be carried out, it will suggest the rough response that is characteristic of most of the room, and for equalization the microphone should be placed in an area with this response. An exception, of course, will be such rooms as recording studio control rooms where there is really only one spot of interest, which must be used for the microphone during equalization.

When an equalization microphone location is chosen, extend the microphone stand and rotate it by the base about this location, with the microphone pointing up. This is shown in Figure 5. If there are sharp changes in response along the circle thus measured, this means there are loudspeaker or local interference patterns very near the chosen measurement position, and another position should be considered.

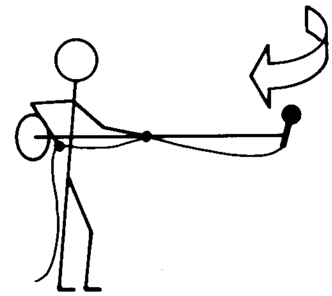
Figure 5. Testing a microphone position for local reflection anomalies.

Measure the ambient noise in the measurement position, and mark it on the screen of the real time analyser. The equalized sound must be at least 6 dB above this noise floor, and preferably more.

Connect the equalizer and the test instrumentation. This has been traditionally done as shown in Figure 6, using either a house microphone or a measurement microphone. Consider using the arrangement shown in Figure 7, developed by Don Davis and published in 1975. First play the pink noise directly through the mixer; then, raise the level on the microphone channel until the VU meter on the mixer is 3 dB above the level of the pink noise alone. This way, the original noise and the regeneration of the noise are at equal levels. The noise signal and its regeneration sound quite different from pink noise alone, without regeneration; it sounds something like a jet engine revving up. It will be necessary to pay attention to levels and not allow the system to run away into feedback. In an alternate arrangement, a house microphone is played into the mixer and an entirely separate measurement microphone, not connected to the mixer, is used with the real time analyser. This equalization technique gives greater gain before feedback than the traditional feedforward-only method, and subjectively better sounding quality for both reinforcement and playback.

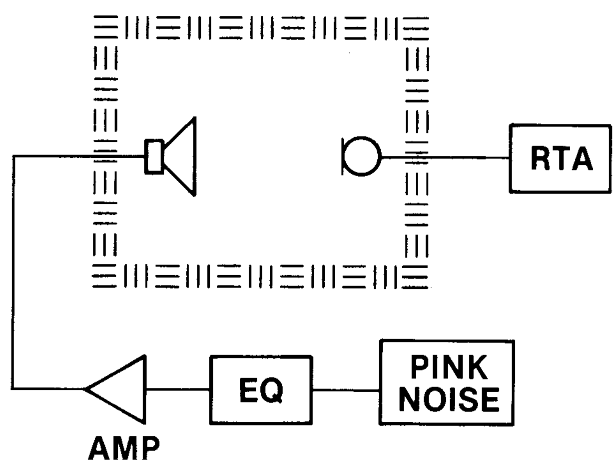

Figure 6. Traditional equalization test set-up.

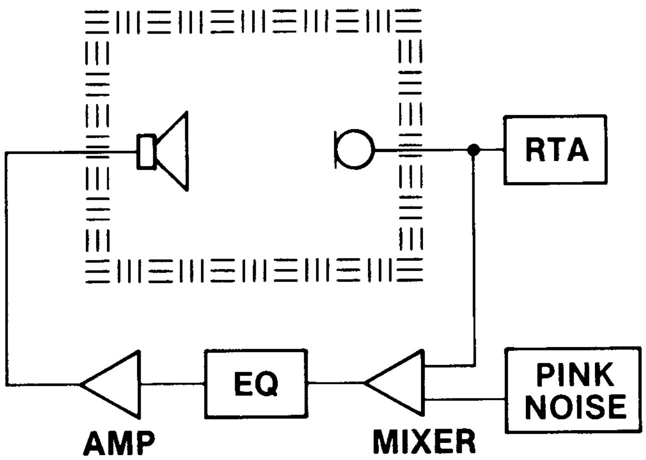

Figure 7. Preferred equalization test set-up with noise and regeneration mixed at equal levels.

Mark the desired house curve on the face of the real time analyser. Adjust the shelving of the crossover network or active crossover so that the raw house curve in about the middle half of the spectrum touches the desired curve at its lowest points. Apply high-pass and low-pass networks before equalizing. Choose them based on your knowledge and experience of the loudspeakers used. Do not attempt to widen the bandwidth of the loudspeakers with electrical equalization; the laws of physics enforce themselves and severely punish those who transgress. If your loudspeakers have a characteristic shelving at their frequency extremes (this is typical of, for example, vented bass horns), alter the desired house curve to reflect this.

Set the analyser to its slow integration time. Move the control for the band highest above the desired curve one or two decibels only, and wait several seconds to see the result on the analyser. If this adjustment is insufficient, and if adjacent bands are also too high, next cut the adjacent bands one or two decibels, rather than continuing to cut the center band.

Continue in this manner, pausing *frequently* to talk through the system (or play recorded material through it) with the experienced listener making a judgment of the system's naturalness. If the naturalness has improved, continue. It is quite possible that the naturalness can get *worse* as the curve on the real time analyser gets *better*; in this case return the equalizer controls to their setting at the last listening test and attempt these corrections more slowly, using more controls moved a smaller amount.

Strive to achieve equalizer settings among which there is no variation from band to band more than three decibels or so, and most variations only one decibel, or two. If you find one band cut more than this beyond its neighbors, try going back and raising that band, or cutting its neighbors, or both, but only a small amount. Remember that the curve on the face of the equalizer should be as smooth as possible, and remember also that two slight dips below the desired response curve are preferable to one slight peak above it, both from a feedback and naturalness point of view.

The least equalization is the best equalization. Using the real time analyser to measure the electrical response of the equalizer is a useful and instructive exercise. The upper curve in Figure 8 shows an equalizer response with a low frequency adjustment making an extreme demand on the power amplifier . . . can the amplifier accept ths demand? It shows a high frequency boost that will enhance transistor hiss and breath noise, and degrade perceived quietness of the system severely . . . can the system signal-to-noise ratio accept this degradation? It shows enormous insertion loss from the filters. Even minimum-phase filters always create phase shifts, greater shifts with greater insertions. Non-minimum-phase filters will introduce much greater shifts than these . . . can naturalness survive this? The lower curve in Figure 8 shows a preferable equalizer response for the same sound system.

Know when to quit.

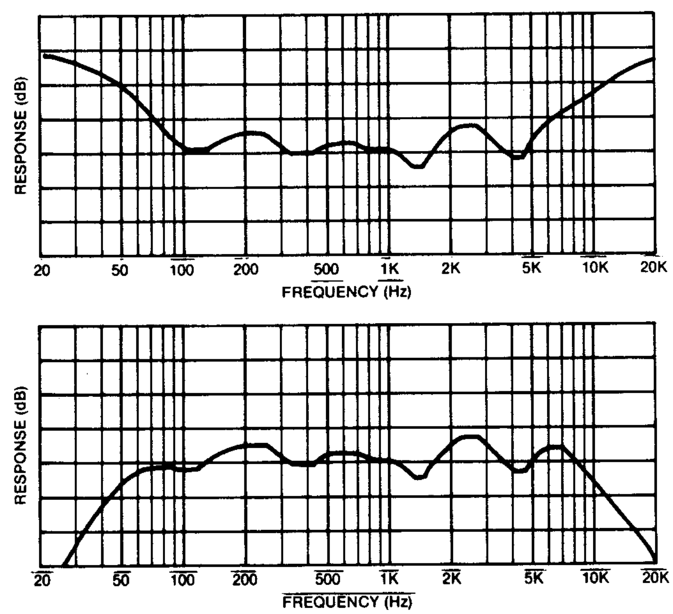

Figure 8. Filter set electrical response with excessive equalization (upper curve) and moderate, preferable equalization (lower curve).

In Conclusion

The most important judges of sound system performance (the users and audiences) will never see a real time analyser, but judge the system on its feedback ceiling and naturalness. It is possible to make these worse while making the real time analyser display better. A true professional has judgment good enough to discern between wheat and chaff.